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This work, which combined the efforts of 10 PIs at 6 different institutions, focused on combining work in small-scale physical oceanography, underwater acoustics, adaptive signal processing, and communications to bridge the gap between the capabilities of current underwater acoustic communications systems and the projected requirements for such systems in U.S. Navy CONOPS for the future. Specifically, the team focused on three main areas. These included: 1) the development of new data modulation and coding, channel estimation, adaptive equalization and decoding, and numerical optimization methods to improve the ability of underwater acoustic communications systems to transmit data reliably through complex ocean environments, 2) characterizing the acoustic propagation properties of those environments and 3) predicting the performance of acoustic communications systems in realistic ocean environments.

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Underwater Acoustic Propagation and Communications: A Coupled Research Program
FINAL REPORT

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1 Abstract

This work, which combined the efforts of 10 PIs at 6 different institutions, focused on combining research in small-scale physical oceanography, underwater acoustics, adaptive signal processing, and communications to bridge the gap between the capabilities of current underwater acoustic communications systems and the projected requirements for such systems in U.S. Navy CONOPS for the future. Specifically, the team focused on three main areas. These included 1) the development of new data modulation and coding, channel estimation, adaptive equalization and decoding, and numerical optimization methods to improve the ability of underwater acoustic communications systems to transmit data reliably through complex ocean environments, 2) characterizing the acoustic propagation properties of those environments and 3) predicting the performance of acoustic communications systems in realistic ocean environments.

2 Project Personnel and Statistics

The MURI project had a significant impact on the training of graduate students and postdoctoral investigators, the state-of-the-art in underwater acoustic communications and the analysis and modeling of the acoustic propagation phenomena which influence it. The funded work resulted in the publication of 46 peer reviewed journal papers and 116 conference papers. Three of the PIs received important professional awards in part based upon their MURI funded research. The papers and investigators funded by this MURI project received additional 15 special recognitions (paper awards, invited, keynote or plenary conference talks, etc.) based in part on their MURI funded work. Finally, 7 postdoctoral investigators and 17 graduate students were supported in whole or in part by the MURI grant. The Principle Investigators funding by this MURI grant and their home institutions are listed below.

Dr. James Preisig (Lead Investigator)	Woods Hole Oceanographic Institution
Dr. Andone Lavery	Woods Hole Oceanographic Institution
Mr. Lee Freitag	Woods Hole Oceanographic Institution
Prof. Arthur Baggeroer	Massachusetts Institute of Technology
Prof. Gregory Wornell	Massachusetts Institute of Technology
Prof. Milica Stojanovic	Northeastern University
Prof. David Farmer	University of Rhode Island
Prof. Andrew Singer	University of Illinois
Dr. Grant Deane	Scripps Institution of Oceanography
Dr. Dale Stokes	Scripps Institution of Oceanography

3 Project Accomplishments

The work pursued and accomplishments achieved with the support of this MURI grant fall into five broad categories. These include:

Modulation and Coding: A number of approaches to modulation and coding were developed, analyzed, and tested/validated with field data. These included

1. a method for rateless random linear packet coding over half-duplex links as well as its variant for links with no feedback This method is suitable for use with any existing physical layer, i.e. with any existing acoustic modem, in point-to-point as well as broadcast scenarios.
2. a modulation methodology referred to as Super Nyquist Modulation wherein the transmitter sends data symbols at a rate that is faster than the "Nyquist" rate for the channel (i.e., the inverse of the channel bandwidth). A companion error correction coding technique which combines a simple base code with dithered repetition coding was developed and shown to be an effective low-complexity method of achieving reliable high-rate communications. When available, feedback from the receiver to the transmitter controls the number or repetitions and creates a rateless coding system that is shown to approach the capacity of the channel.
3. a joint modulation/coding technique suitable for both SIMO and MIMO systems.
4. an adaptive OFDM modulation technique, whereby the transmitter acts in response to (delayed) feedback from the receiver, enabling it to adapt to environmental fluctuations.
5. a transmit diversity technique based on Alamouti space-frequency coding. The technique is applicable in both coherent and differentially coherent forms.

Channel Estimation, Equalization and Decoding: Equalizers which reduce the impact of ambient noise and propagation induced distortions were one of the major areas of work in this MURI. The specific approaches ranged from the initial channel estimation that is a part of many equalizers adaptation to environmental fluctuations to the adaptation algorithms themselves and on to decoding algorithms to correct error using the properties of the signal's error correction codes and are listed below. All approaches and algorithms were tested with field data.

1. Adaptive equalization/decoding techniques using both direct-adaptation and channel estimate based adaptation for SIMO and MIMO systems in a interactive turbo-equalization framework were developed and analyzed. MIMO and SISO transmissions yielded data rates in excess of 30kbps over time-varying channel conditions with mobile receivers at platform speeds of up to a few knots in challenging environments.
2. A class of signal processing techniques for multi-carrier signal detection, whose key innovation is a pre-fitering method for reducing the Doppler distortion along with dedicated synchronization were developed. The methods, termed "multiple-FFT demodulation", were demonstrated in a variety of experimental conditions (including mobile settings) and over extended periods of time (days), showing very robust performance.
3. A frequency domain equalization algorithm for underwater acoustic communications was developed. While frequency domain equalization has been considered for other environments in the past, the technique developed here improves the ability to track channel fluctuations, reduces computational complexity, and improves performance in the low

SNR regime when compared to previously techniques. The algorithm is particularly advantageous in high data rate systems employing Super Nyquist Modulation.

4. An equalizer adaptation algorithm suitable for modern turbo equalizers which reduces computational complexity and improves overall performance in rapidly time-varying environments such as the underwater acoustic communications channel was developed. Turbo equalization algorithms, while computationally expensive, offer great potential for achieving reliable high-rate communications in difficult environments. The algorithm developed here is known as Recursive Expected Least Squares (RELS). It is a decision directed adaptive equalization algorithm that exploits soft (i.e., probabilistic) information in a unique Expectation-Maximize (EM) framework both in the feedback filter of a decision feedback equalizer and in the equalizer filter adaptation process itself. The RELS algorithms showed superior equalization performance when compared to hard decision directed equalizers in non-iterative equalizers and more rapid convergence in turbo-equalization receivers than the previous state-of-the-art systems. In low SNR situations, the new soft-decision directed algorithms were able to converge to meaningful solutions while the previous state-of-the-art systems were not even able to converge.
5. A new channel estimation algorithm suitable for environments in which the statistical fluctuations of the intensity of individual multi-path arrivals have "heavy tails" (for example, are characterized by a k-distribution) was developed and demonstrated. (See item 2 under **Acoustic Propagation** below.)
6. In many environments, the underwater acoustic communications channel exhibits a sparse structure in either the impulse response or the scattering function domain. An optimization algorithm that uses a weighted linear combination of an L1 and and L2 norm that achieves better results at a lower computational complexity than previous mixed-norm algorithms was developed. Applying the optimization algorithm to the estimation of the underwater acoustic communications channel, we achieved performance that is better than that of the traditional Least Squares (L2 norm) approach to channel estimation.
7. Least squares adaptation algorithms utilize sample correlation matrices of the received signal to calculate optimal filter coefficients. A new technique of estimating the sample correlation matrix of signals received at a linear vertical array was developed. The technique, which assumes a Toeplitz structure of the ensemble correlation matrix and forces the sample correlation matrix to have the same structure, was shown to reduce computational complexity and improve equalizer adaptation performance.
8. Beamspace adaptive equalization techniques were developed, analyzed, and tested. The techniques exploited the limited extent of the vertical wavenumber spectrum of signals propagating in a waveguide such as the ocean to improve performance of the resulting equalization algorithms.

Acoustic Propagation: The analysis and characterization of the propagation of acoustic signals in the ocean environment is critical of two reasons. First, it allows the performance of acoustic communications systems as a function of environmental conditions to be predicted and modulation and demodulation techniques and communication system configurations to be optimized to maximize performance. Thus, the acoustic propagation work under this MURI program as described below was undertaken to couple the analysis of acoustic propagation to the development of algorithms and the optimization of system configurations.

1. A statistical channel model and accompanying simulator (available online at http://millitsa.coe.neu.edu) were developed. The model captures large and small-scale variations (random displacements over many/few wavelengths), random Doppler shifting and multipath, and can be used for analyses of capacity, BER, outage, coverage, etc.
2. Surface Wave Focusing is an acoustic propagation phenomena in which the curvature of the ocean surface focuses surface scattered acoustic signals in the same manner as would a curved mirror. In many environments, this results in very high intensity and rapid fluctuations in the amplitude of surface reflected signals with the probability distribution of the arrival intensity termed "heavy-tailed". Such fluctuations are particularly disruptive to the reliable operation of equalization algorithms. Extensive work was done characterizing the deterministic physics of the formation and propagation of surface wave focused signals and their statistical characterization.

Performance Prediction and System Optimization: The vast majority of adaptive equalization algorithms rely on sample correlation matrices of the received signal to calculate optimal equalizer filter coefficients. This presents a huge challenge to both reliable adaptation and the prediction of equalizer performance when the number of observations available to calculate the sample correlation matrix is approximately equal to or less than the number of equalizer coefficients that need to be estimated. This situation is referred to as the "low snapshot" regime. A method based upon Asymptotic Random Matrix Theory (RMT) to predict the performance of Least Squares channel estimation and equalization algorithms in rapidly time-varying environments was developed. Three applications of the RMT technique were pursued as detailed below.

1. The impact of diagonal loading on adaptive array processing algorithms (MPDR) in the low snapshot regime was analyzed and guidelines for adaptive diagonal loading were developed.
2. The performance of least squares channel estimation algorithms as a function of number of observations was analyzed and new insights gains about the performance of algorithms as the number of available observations changes from greater than to less than the number of coefficients that must be estimated.
3. The RMT technique was used to analyze and optimize the choice of number of and spacing between the hydrophones of a linear array used for multi-channel equalization and the filter length of the corresponding multi-channel adaptive equalizers. The work extended traditional theory to include the effects of finite and limited observation intervals over which data can be gathered with which the equalizer filter coefficients can be optimized and led to fundamental new insights into the impact of propagation physics on optimal equalizer configuration.

Field Work: Over the duration of the MURI grant, five major field experiments were conducted with full or partial support from the grant. These included the RACE08, SPACE08, MACE10, NoisEx10 and KAM11 field experiments. These experiments collected field data to support the research of the MURI team as well as numerous underwater acoustic communications investigators on other ONR funded programs. The data was also used to benchmark adaptive equalization algorithms running on existing underwater acoustic communications modems as a point of comparison for the new algorithms developed under the MURI program.

4 Applicability to Navy Needs

The algorithms developed and knowledge of underwater acoustic propagation and communications performance prediction gained are directly applicable to future needs of the Navy. Taken as a body of work, the modulation, coding, demodulation, and decoding algorithms are a significant reservoir of techniques available for transition to current or new Navy systems at a time when transition funding is available to do so. This transition would improve upon the capabilities of the current generation of underwater acoustic communications systems. The acoustic propagation analysis and results will be able to continue to support the development and analysis of acoustic communications systems. The performance prediction results, with additional funding, will be able to support operational decision tools as well as the optimization of underwater acoustic communications algorithms and systems into the future.